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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
10/078,441	02/21/2002	Naoshi Matsuo	1359.1062	4915

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EXAMINER	
HAROLD, JEFFEREY F	
ART UNIT	PAPER NUMBER
2644	4

DATE MAILED: 06/07/2004

Please find below and/or attached an Office communication concerning this application or proceeding.

Office Action Summary	Application No.	Applicant(s)
	10/078,441	MATSUO, NAOSHI
	Examiner	Art Unit
	Jefferey F Harold	2644

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on 21 February 2002.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) 1-21 is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) 1-21 is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☒ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☒ All b)☐ Some * c)☐ None of:

1.☒ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☒ Information Disclosure Statement(s) (PTO-1449 or PTO/SB/08) Paper No(s)/Mail Date 3.

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .

5) ☐ Notice of Informal Patent Application (PTO-152)

6) ☐ Other: _____.

DETAILED ACTION

Information Disclosure Statement

1. The references listed in the Information Disclosure Statement submitted on April 25, 2002, have been considered by the examiner (see attached PTO-1449).

Double Patenting

The nonstatutory double patenting rejection is based on a judicially created doctrine grounded in public policy (a policy reflected in the statute) so as to prevent the unjustified or improper timewise extension of the "right to exclude" granted by a patent and to prevent possible harassment by multiple assignees. See *In re Goodman*, 11 F.3d 1046, 29 USPQ2d 2010 (Fed. Cir. 1993); *In re Longi*, 759 F.2d 887, 225 USPQ 645 (Fed. Cir. 1985); *In re Van Ornum*, 686 F.2d 937, 214 USPQ 761 (CCPA 1982); *In re Vogel*, 422 F.2d 438, 164 USPQ 619 (CCPA 1970); and, *In re Thorington*, 418 F.2d 528, 163 USPQ 644 (CCPA 1969).

A timely filed terminal disclaimer in compliance with 37 CFR 1.321(c) may be used to overcome an actual or provisional rejection based on a nonstatutory double patenting ground provided the conflicting application or patent is shown to be commonly owned with this application. See 37 CFR 1.130(b).

Effective January 1, 1994, a registered attorney or agent of record may sign a terminal disclaimer. A terminal disclaimer signed by the assignee must fully comply with 37 CFR 3.73(b).

2. ***Claims 1-21*** are rejected under the judicially created doctrine of obviousness-type double patenting as being unpatentable over claims 1-3 of U.S. Patent No. 6,317,501, hereinafter referenced as '501. Although the conflicting claims are not identical, they are not patentably distinct from each other because both the instant application and the patent utilize the microphone array to determine filter coefficients for the echo cancellation process. However, the instant application provides claims that are further limiting wherein more details are provided concerning the echo cancellation process.

Regarding **claim 1**, '501 discloses a microphone array apparatus. In addition '501 discloses an echo cancellation processing system comprising in an inherent full duplex telephony system: a microphone array; a loudspeaker for converting a speech signal transmitted from a telephony system on a communication partner side to a speech; and an echo cancellation processing part comprising an estimated wraparound speech signal generating part for estimating a speech signal that is outputted from the loudspeaker and wraps around to the microphone array, using a time difference or a level difference between input speech signals of a plurality of channels of the microphone array, and generate an estimated wraparound speech signal in accordance with an estimated result based on an output speech signal supplied to the loudspeaker, and a subtracter for subtracting the estimated wraparound speech signal from an input speech signal inputted to the microphone array, as disclosed in claims 1-3.

Regarding **claim 2**, '501discloses everything claimed as applied above (see claim 1), in addition, '501 discloses a wraparound delay amount detecting part for comparing an output speech signal supplied to the loudspeaker with a wraparound speech signal contained in an input speech signal inputted through the microphone array, and detecting a delay amount of the wraparound speech signal contained in the input speech signal delayed from the output speech signal; and a delay processing part for delaying the output speech signal in accordance with the delay amount detected by the wraparound delay amount detecting part, wherein an output speech signal of the delay processing part is inputted to the estimated wraparound speech signal generating part as an input signal, as disclosed in claims 1-3.

Regarding **claim 3**, '501 discloses everything claimed as applied above (see claim 2), in addition, '501 discloses a wraparound speech signal emphasizing part for emphasizing and extracting the wraparound speech signal from the input speech signal, wherein the wraparound speech signal emphasizing part comprises: a first delay amount calculating part for calculating a delay amount between the respective microphone input signals delayed from the loudspeaker based on input speech signals inputted through each microphone constituting the microphone array; and a first addition processing part for conducting synchronous addition processing regarding an input speech signal inputted through each microphone constituting the microphone array, by adjusting the delay amount between the respective microphone input signals delayed from the loudspeaker, and emphasizing the wraparound speech signal, and the emphasized wraparound speech signal is inputted to the wraparound delay amount detecting part, as disclosed in claims 1-3.

Regarding **claim 8**, '501 discloses everything claimed as applied above (see claim 3), in addition, '501 discloses a speaker's speech signal emphasizing part for conducting synchronous addition processing of a speaker's speech signal inputted through each microphone constituting the microphone array, and emphasizing the speaker's speech signal, thereby generating an input speech signal in which a speaker signal is emphasized, as disclosed in claims 1-3.

Regarding **claim 11**, '501 discloses everything claimed as applied above (see claim 2), in addition, '501 discloses a speaker's speech signal emphasizing part for conducting synchronous addition processing of a speaker's speech signal inputted

through each microphone constituting the microphone array, and emphasizing the speaker's speech signal, thereby generating an input speech signal in which a speaker signal is emphasized, as disclosed in claims 1-3.

Regarding **claim 14**, '501 discloses everything claimed as applied above (see claim 1), in addition, '501 discloses a speaker's speech signal emphasizing part for conducting synchronous addition processing of a speaker's speech signal inputted through each microphone constituting the microphone array, and emphasizing the speaker's speech signal, thereby generating an input speech signal in which a speaker signal is emphasized, as disclosed in claims 1-3.

Regarding **claim 17**, '501 discloses everything claimed as applied above (see claim 1), in addition, '501 discloses wherein the estimated wraparound speech signal generating part comprises an adaptive filter, and a coefficient updating part for updating a coefficient of the adaptive filter at a predetermined timing, wherein the coefficient updating part determines the estimated result and a coefficient update amount of the adaptive filter based on a level of a wraparound speech signal remaining in an echo cancellation result obtained by the echo cancellation processing part, and the adaptive filter conducts the adaptation based on an output speech signal supplied to the loudspeaker and generates the estimated wraparound speech signal, as disclosed in claims 1-3.

Regarding **claim 21**, '501 discloses a recording medium storing a processing program of a full duplex telephony system, the program comprising: a processing operation of controlling a microphone array in which a plurality of microphones are

disposed at predetermined positions; a processing operation of controlling a loudspeaker for converting a speech signal transmitted from a telephony system on a communication partner side to a speech signal; and an echo cancellation processing operation comprising an estimated wraparound speech signal generation processing operation of estimating a speech signal that is outputted from the loudspeaker and wraps around to the microphone array, using an input speech signal of the microphone array, and generating an estimated wraparound speech signal in accordance with an estimated result based on an output speech signal supplied to the loudspeaker, and a subtraction processing operation of subtracting the estimated wraparound speech signal from an input speech signal inputted through the microphone array.

3. ***Claims 1-21*** are provisionally rejected under the judicially created doctrine of obviousness-type double patenting as being unpatentable over claims 1-3 of copending Application No. 10/003,768, hereinafter referenced as '768. Although the conflicting claims are not identical, they are not patentably distinct from each other because both the instant application and the patent utilize the microphone array to determine filter coefficients for the echo cancellation process. However, the instant application provides claims that are further limiting wherein more details are provided concerning the echo cancellation process.

Regarding **claim 1**, '768 discloses a microphone array apparatus. In addition '768 discloses an echo cancellation processing system comprising in an inherent full duplex telephony system: a microphone array; a loudspeaker for converting a speech

signal transmitted from a telephony system on a communication partner side to a speech; and an echo cancellation processing part comprising an estimated wraparound speech signal generating part for estimating a speech signal that is outputted from the loudspeaker and wraps around to the microphone array, using a time difference or a level difference between input speech signals of a plurality of channels of the microphone array, and generate an estimated wraparound speech signal in accordance with an estimated result based on an output speech signal supplied to the loudspeaker, and a subtracter for subtracting the estimated wraparound speech signal from an input speech signal inputted to the microphone array, as disclosed in claims 1-3.

Regarding **claim 2**, '768 discloses everything claimed as applied above (see claim 1), in addition, '768 discloses a wraparound delay amount detecting part for comparing an output speech signal supplied to the loudspeaker with a wraparound speech signal contained in an input speech signal inputted through the microphone array, and detecting a delay amount of the wraparound speech signal contained in the input speech signal delayed from the output speech signal; and a delay processing part for delaying the output speech signal in accordance with the delay amount detected by the wraparound delay amount detecting part, wherein an output speech signal of the delay processing part is inputted to the estimated wraparound speech signal generating part as an input signal, as disclosed in claims 1-3.

Regarding **claim 3**, '768 discloses everything claimed as applied above (see claim 2), in addition, '768 discloses a wraparound speech signal emphasizing part for emphasizing and extracting the wraparound speech signal from the input speech signal,

wherein the wraparound speech signal emphasizing part comprises: a first delay amount calculating part for calculating a delay amount between the respective microphone input signals delayed from the loudspeaker based on input speech signals inputted through each microphone constituting the microphone array; and a first addition processing part for conducting synchronous addition processing regarding an input speech signal inputted through each microphone constituting the microphone array, by adjusting the delay amount between the respective microphone input signals delayed from the loudspeaker, and emphasizing the wraparound speech signal, and the emphasized wraparound speech signal is inputted to the wraparound delay amount detecting part, as disclosed in claims 1-3.

Regarding **claim 8**, '768 discloses everything claimed as applied above (see claim 3), in addition, '768 discloses a speaker's speech signal emphasizing part for conducting synchronous addition processing of a speaker's speech signal inputted through each microphone constituting the microphone array, and emphasizing the speaker's speech signal, thereby generating an input speech signal in which a speaker signal is emphasized, as disclosed in claims 1-3.

Regarding **claim 11**, '768 discloses everything claimed as applied above (see claim 2), in addition, '768 discloses a speaker's speech signal emphasizing part for conducting synchronous addition processing of a speaker's speech signal inputted through each microphone constituting the microphone array, and emphasizing the speaker's speech signal, thereby generating an input speech signal in which a speaker signal is emphasized, as disclosed in claims 1-3.

Regarding **claim 14**, '768 discloses everything claimed as applied above (see claim 1), in addition, '768 discloses a speaker's speech signal emphasizing part for conducting synchronous addition processing of a speaker's speech signal inputted through each microphone constituting the microphone array, and emphasizing the speaker's speech signal, thereby generating an input speech signal in which a speaker signal is emphasized, as disclosed in claims 1-3.

Regarding **claim 17**, '501 discloses everything claimed as applied above (see claim 1), in addition, '768 discloses wherein the estimated wraparound speech signal generating part comprises an adaptive filter, and a coefficient updating part for updating a coefficient of the adaptive filter at a predetermined timing, wherein the coefficient updating part determines the estimated result and a coefficient update amount of the adaptive filter based on a level of a wraparound speech signal remaining in an echo cancellation result obtained by the echo cancellation processing part, and the adaptive filter conducts the adaptation based on an output speech signal supplied to the loudspeaker and generates the estimated wraparound speech signal, as disclosed in claims 1-3.

Regarding **claim 21**, '768 discloses a recording medium storing a processing program of a full duplex telephony system, the program comprising: a processing operation of controlling a microphone array in which a plurality of microphones are disposed at predetermined positions; a processing operation of controlling a loudspeaker for converting a speech signal transmitted from a telephony system on a communication partner side to a speech signal; and an echo cancellation processing

operation comprising an estimated wraparound speech signal generation processing operation of estimating a speech signal that is outputted from the loudspeaker and wraps around to the microphone array, using an input speech signal of the microphone array, and generating an estimated wraparound speech signal in accordance with an estimated result based on an output speech signal supplied to the loudspeaker, and a subtraction processing operation of subtracting the estimated wraparound speech signal from an input speech signal inputted through the microphone array.

This is a provisional obviousness-type double patenting rejection because the conflicting claims have not in fact been patented.

4. ***Claims 1-21*** are provisionally rejected under the judicially created doctrine of obviousness-type double patenting as being unpatentable over claims 1-3 of copending Application No. 10/035,507, hereinafter referenced as '507. Although the conflicting claims are not identical, they are not patentably distinct from each other because both the instant application and the patent utilize the microphone array to determine filter coefficients for the echo cancellation process. However, the instant application provides claims that are further limiting wherein more details are provided concerning the echo cancellation process.

Regarding **claim 1**, '507 discloses a microphone array apparatus. In addition '507 discloses an echo cancellation processing system comprising in an inherent full duplex telephony system: a microphone array; a loudspeaker for converting a speech signal transmitted from a telephony system on a communication partner side to a speech; and an echo cancellation processing part comprising an estimated wraparound

speech signal generating part for estimating a speech signal that is outputted from the loudspeaker and wraps around to the microphone array, using a time difference or a level difference between input speech signals of a plurality of channels of the microphone array, and generate an estimated wraparound speech signal in accordance with an estimated result based on an output speech signal supplied to the loudspeaker, and a subtracter for subtracting the estimated wraparound speech signal from an input speech signal inputted to the microphone array, as disclosed in claims 1-3.

Regarding **claim 2**, '507 discloses everything claimed as applied above (see claim 1), in addition, '507 discloses a wraparound delay amount detecting part for comparing an output speech signal supplied to the loudspeaker with a wraparound speech signal contained in an input speech signal inputted through the microphone array, and detecting a delay amount of the wraparound speech signal contained in the input speech signal delayed from the output speech signal; and a delay processing part for delaying the output speech signal in accordance with the delay amount detected by the wraparound delay amount detecting part, wherein an output speech signal of the delay processing part is inputted to the estimated wraparound speech signal generating part as an input signal, as disclosed in claims 1-3.

Regarding **claim 3**, '507 discloses everything claimed as applied above (see claim 2), in addition, '507 discloses a wraparound speech signal emphasizing part for emphasizing and extracting the wraparound speech signal from the input speech signal, wherein the wraparound speech signal emphasizing part comprises: a first delay amount calculating part for calculating a delay amount between the respective

microphone input signals delayed from the loudspeaker based on input speech signals inputted through each microphone constituting the microphone array; and a first addition processing part for conducting synchronous addition processing regarding an input speech signal inputted through each microphone constituting the microphone array, by adjusting the delay amount between the respective microphone input signals delayed from the loudspeaker, and emphasizing the wraparound speech signal, and the emphasized wraparound speech signal is inputted to the wraparound delay amount detecting part, as disclosed in claims 1-3.

Regarding **claim 8**, '507 discloses everything claimed as applied above (see claim 3), in addition, '507 discloses a speaker's speech signal emphasizing part for conducting synchronous addition processing of a speaker's speech signal inputted through each microphone constituting the microphone array, and emphasizing the speaker's speech signal, thereby generating an input speech signal in which a speaker signal is emphasized, as disclosed in claims 1-3.

Regarding **claim 11**, '507 discloses everything claimed as applied above (see claim 2), in addition, '507 discloses a speaker's speech signal emphasizing part for conducting synchronous addition processing of a speaker's speech signal inputted through each microphone constituting the microphone array, and emphasizing the speaker's speech signal, thereby generating an input speech signal in which a speaker signal is emphasized, as disclosed in claims 1-3.

Regarding **claim 14**, '507 discloses everything claimed as applied above (see claim 1), in addition, '507 discloses a speaker's speech signal emphasizing part for

conducting synchronous addition processing of a speaker's speech signal inputted through each microphone constituting the microphone array, and emphasizing the speaker's speech signal, thereby generating an input speech signal in which a speaker signal is emphasized, as disclosed in claims 1-3.

Regarding **claim 17**, '501 discloses everything claimed as applied above (see claim 1), in addition, '507 discloses wherein the estimated wraparound speech signal generating part comprises an adaptive filter, and a coefficient updating part for updating a coefficient of the adaptive filter at a predetermined timing, wherein the coefficient updating part determines the estimated result and a coefficient update amount of the adaptive filter based on a level of a wraparound speech signal remaining in an echo cancellation result obtained by the echo cancellation processing part, and the adaptive filter conducts the adaptation based on an output speech signal supplied to the loudspeaker and generates the estimated wraparound speech signal, as disclosed in claims 1-3.

Regarding **claim 21**, '507 discloses a recording medium storing a processing program of a full duplex telephony system, the program comprising: a processing operation of controlling a microphone array in which a plurality of microphones are disposed at predetermined positions; a processing operation of controlling a loudspeaker for converting a speech signal transmitted from a telephony system on a communication partner side to a speech signal; and an echo cancellation processing operation comprising an estimated wraparound speech signal generation processing operation of estimating a speech signal that is outputted from the loudspeaker and

wraps around to the microphone array, using an input speech signal of the microphone array, and generating an estimated wraparound speech signal in accordance with an estimated result based on an output speech signal supplied to the loudspeaker, and a subtraction processing operation of subtracting the estimated wraparound speech signal from an input speech signal inputted through the microphone array.

This is a provisional obviousness-type double patenting rejection because the conflicting claims have not in fact been patented.

5. ***Claims 1-21*** are provisionally rejected under the judicially created doctrine of obviousness-type double patenting as being unpatentable over claims 1-3 of copending Application No. 10/038,188, hereinafter referenced as '188. Although the conflicting claims are not identical, they are not patentably distinct from each other because both the instant application and the patent utilize the microphone array to determine filter coefficients for the echo cancellation process. However, the instant application provides claims that are further limiting wherein more details are provided concerning the echo cancellation process.

Regarding **claim 1**, '188 discloses a microphone array apparatus. In addition '188 discloses an echo cancellation processing system comprising in an inherent full duplex telephony system: a microphone array; a loudspeaker for converting a speech signal transmitted from a telephony system on a communication partner side to a speech; and an echo cancellation processing part comprising an estimated wraparound speech signal generating part for estimating a speech signal that is outputted from the loudspeaker and wraps around to the microphone array, using a time difference or a

level difference between input speech signals of a plurality of channels of the microphone array, and generate an estimated wraparound speech signal in accordance with an estimated result based on an output speech signal supplied to the loudspeaker, and a subtracter for subtracting the estimated wraparound speech signal from an input speech signal inputted to the microphone array, as disclosed in claims 1-3.

Regarding **claim 2**, '188 discloses everything claimed as applied above (see claim 1), in addition, '188 discloses a wraparound delay amount detecting part for comparing an output speech signal supplied to the loudspeaker with a wraparound speech signal contained in an input speech signal inputted through the microphone array, and detecting a delay amount of the wraparound speech signal contained in the input speech signal delayed from the output speech signal; and a delay processing part for delaying the output speech signal in accordance with the delay amount detected by the wraparound delay amount detecting part, wherein an output speech signal of the delay processing part is inputted to the estimated wraparound speech signal generating part as an input signal, as disclosed in claims 1-3.

Regarding **claim 3**, '188 discloses everything claimed as applied above (see claim 2), in addition, '188 discloses a wraparound speech signal emphasizing part for emphasizing and extracting the wraparound speech signal from the input speech signal, wherein the wraparound speech signal emphasizing part comprises: a first delay amount calculating part for calculating a delay amount between the respective microphone input signals delayed from the loudspeaker based on input speech signals inputted through each microphone constituting the microphone array; and a first addition

processing part for conducting synchronous addition processing regarding an input speech signal inputted through each microphone constituting the microphone array, by adjusting the delay amount between the respective microphone input signals delayed from the loudspeaker, and emphasizing the wraparound speech signal, and the emphasized wraparound speech signal is inputted to the wraparound delay amount detecting part, as disclosed in claims 1-3.

Regarding **claim 8**, '188 discloses everything claimed as applied above (see claim 3), in addition, '188 discloses a speaker's speech signal emphasizing part for conducting synchronous addition processing of a speaker's speech signal inputted through each microphone constituting the microphone array, and emphasizing the speaker's speech signal, thereby generating an input speech signal in which a speaker signal is emphasized, as disclosed in claims 1-3.

Regarding **claim 11**, '188 discloses everything claimed as applied above (see claim 2), in addition, '188 discloses a speaker's speech signal emphasizing part for conducting synchronous addition processing of a speaker's speech signal inputted through each microphone constituting the microphone array, and emphasizing the speaker's speech signal, thereby generating an input speech signal in which a speaker signal is emphasized, as disclosed in claims 1-3.

Regarding **claim 14**, '188 discloses everything claimed as applied above (see claim 1), in addition, '188 discloses a speaker's speech signal emphasizing part for conducting synchronous addition processing of a speaker's speech signal inputted through each microphone constituting the microphone array, and emphasizing the

speaker's speech signal, thereby generating an input speech signal in which a speaker signal is emphasized, as disclosed in claims 1-3.

Regarding **claim 17**, '501 discloses everything claimed as applied above (see claim 1), in addition, '188 discloses wherein the estimated wraparound speech signal generating part comprises an adaptive filter, and a coefficient updating part for updating a coefficient of the adaptive filter at a predetermined timing, wherein the coefficient updating part determines the estimated result and a coefficient update amount of the adaptive filter based on a level of a wraparound speech signal remaining in an echo cancellation result obtained by the echo cancellation processing part, and the adaptive filter conducts the adaptation based on an output speech signal supplied to the loudspeaker and generates the estimated wraparound speech signal, as disclosed in claims 1-3.

Regarding **claim 21**, '188 discloses a recording medium storing a processing program of a full duplex telephony system, the program comprising: a processing operation of controlling a microphone array in which a plurality of microphones are disposed at predetermined positions; a processing operation of controlling a loudspeaker for converting a speech signal transmitted from a telephony system on a communication partner side to a speech signal; and an echo cancellation processing operation comprising an estimated wraparound speech signal generation processing operation of estimating a speech signal that is outputted from the loudspeaker and wraps around to the microphone array, using an input speech signal of the microphone array, and generating an estimated wraparound speech signal in accordance with an

estimated result based on an output speech signal supplied to the loudspeaker, and a subtraction processing operation of subtracting the estimated wraparound speech signal from an input speech signal inputted through the microphone array.

This is a provisional obviousness-type double patenting rejection because the conflicting claims have not in fact been patented.

Regarding **claims 4-7, 9, 10, 12, 13, 15, 16, and 18-20**, they are rejected because they depend from the above rejected claims.

Conclusion

5. Any inquiry concerning this communication or earlier communications from the examiner should be directed to Jefferey F Harold whose telephone number is 703-306-5836. The examiner can normally be reached on Monday - Friday 9 am - 5:30 pm.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Forester W Isen can be reached on 703-305-4386. The fax phone number for the organization where this application or proceeding is assigned is 703-872-9306.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free).

JFH
May 26, 2004

Jefferey F Harold
Examiner
Art Unit 2644